TYPES OF A. F. AMPLIFIERS

AMPLIFIER CLASSES

LESSON NO. ND-16

SPRAYBERRY ACADEMY of RADIO

ESTABLISHED 1932

PUEBLO, COLORADO

TYPES OF A. F. AMPLIFIERS

AMPLIFIER CLASSES

LESSON ND-16

From your study of the previous lesson you have learned how an A.F. amplifier stage or stages accepts a demodulated signal from a detector tube and builds it up to usable strength by means of amplification. This, in itself, is a remarkable achievement, but it still leaves much to be desired from a *quality* viewpoint. While it is true that the A.F. amplifiers previously described provide amplification, yet, at the same time, they also build up, along with the signal, other undesirable factors which do not make for good speaker reproduction. *The speaker, of course, reproduces what is fed to it,* so if there are undesirable qualities in the electrical equivalent of the sound, they must be omitted or reduced to a minimum before they reach the speaker. All of these undesirable qualities can be summed up in just one word, *distortion.*

Now distortion can be due to anything which permits the output signal of a tube to be different than the input signal with the exception, of course, of normal amplification. The one desirable thing we want from an amplifier stage is *true linear amplification* with the output appearing as an *exact magnified copy of the input.* We don't get this from a single tube in an amplifier stage even if it is operated under ideal conditions. All amplifiers have imperfections because no such device can be 100% or more perfect. Thus, some compromises are always necessary. The type of amplification which has the least imperfections and forces us to accept the least compromises is the one which results in the least distortion. That type, therefore, is the ideal for which acoustic engineers have striven over the years. Results show that remarkable progress has been made.

This matter of perfection in amplification must, however, be tempered with good judgment. No final, lasting good is achieved in a practical way if improvements in quality cannot be detected by the human ear. It is a fact that few people have ears sensitive enough to appreciate the difference between good and bad quality reproduction from a radio set. Other people can detect the difference instantly. Thus, practical radio men must be prepared to service and repair both good and bad quality receivers. It is not the serviceman's function to design radio sets but rather to restore them to their original manufactured condition, be they good or bad from an engineering viewpoint. Beyond that, the repairman should not attempt to go.

By way of introduction, it is necessary to get ahead of the main story of this lesson. There is a type of audio amplification which overcomes much of the distortion as previously mentioned. It utilizes two tubes working

together, each cancelling out the bad effects of the other to give more desirable amplification. It is called *push-pull* amplification, the details of which will be taken up further on in this lesson. The need for push-pull amplification is established from the fact that no single tube of itself has a perfectly straight Eg-Ip characteristic. No matter where along the Eg-Ip curve the operating point is placed, there still will be slight curvature even on the most choice point along the curve. Even with the push-pull arrangement there is still the same curvature of the Eg-Ip curve as with single tubes, but because of the action of the circuit, the effect of this is cancelled out to a large degree, as will be brought out later.

As already mentioned, most of the undesirable qualities of vacuum tube amplification all come under the head of *distortion.* This is a rather broad term and needs further definition. So distortion will be further defined and described, and then we will show how push-pull amplification can be used to keep it to a minimum.

It has been established that as long as the *total distortion* is below 5% of the entire output of a radio set, it will not be objectionable. This should be kept in mind as you study about distortion. In low frequency amplifiers, there are three types of distortion. These are: (1) *frequency distortion,* (2) amplitude or harmonic distortion and (3) phase distortion. The first two are important in sound reproduction while the third is more important in television reproduction. Thus, a discussion of the latter will be left to your television lessons where it has the greatest application.

Frequency distortion is present when an A.F. amplifier does not amplify equally all frequencies fed to it. A certain amount of this type of distortion is, therefore, present in all A.F. amplifiers. If the A.F. frequency range is from 30 to 10,000 cycles, not every frequency in that range is going to be amplified alike. If this were true, a graph of the A.F. output of a radio set would be a *straight line* (linear) from 30 to 10,000 cycles. Few radio sets, indeed, would show this characteristic. Actually, the curve of the output of the average radio set would look somewhat like that shown in Fig. 1. Note the curve starts to dip below 80 cycles and above 4,500 cycles. It is at these points where *frequency distortion* starts, and the more the curve drops from the horizontal at the ends, the more the distortion. Note Fig. 1 employs a logarithmic scale which permits showing the entire output curve in a small space. In general, it may be said that frequency distortion is caused more by circuit parts than tubes. This is because such parts show or present varying characteristics as the frequency changes. For instance, you know that as the frequency changes the impedance of most parts change and, therefore, the voltage drop across these parts changes which results in a change of gain at the tube. A certain amount of this cannot, therefore, be avoided.

Next is *amplitude distortion,* and this really plays havoc with the sound waves as finally emitted from the speaker. This is the distortion that is most distasteful to the ear, and design engineers strive to keep it to a minimum in a given receiver. In order to understand this type of distortion, *it is necessary to have a knowledge of the wave forms which are the electrical equivalents of the actual sound waves.* This is most important because the student must realize that tones

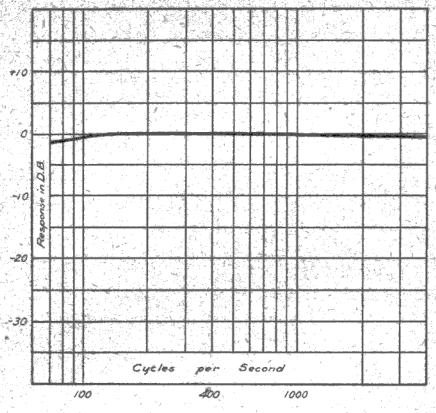

FIG.1

which we recognize as sounds cannot and should not be regarded as *simple cycles* such as you have been accustomed to studying up to now. Your conception of a cycle is probably like the wave form shown in Fig. 2, and it is correct, but do not make the mistake of thinking that this is representative of a *tone wave form* such as that of musical notes or speech. *Tones are not made up of single cycles or wave forms.* They are *much more complex,* being made up of several wave forms and you should remember that the electrical equivalents of these have similar wave forms—that is, a point you should keep in mind at all times.

FUNDAMENTALS OF SOUND

This lesson is not concerned with the actual production of sound waves as that will be left to a later lesson. It is, however, concerned with the electrical equivalents of sound. Thus, in the following remember that reference is made to equivalent electrical wave forms which will cause a speaker to produce actual sound waves.

First consider the degree of loudness or of volume. The amplitude of the wave form (its relative height) determines the level of volume. If the amplitude is small the volume will be low, and if the amplitude is great the volume will be high. Thus a radio volume control controls the amplitude of the A.F. wave form. Volume level is usually measured in units of the decibel or db as explained in your S.A.R. Math Book.

Another unit of sound is its *pitch* as musicians call it. A musical note is said to be low, medium or high in pitch. This refers to its fundamental frequency. The following table which is based on the international pitch gives the fundamental frequency range of musical instruments and of the human voice.

Source	*Frequency in Cycles*
Organ	32-4136
Piano	26-4138
Violin	194-3284
Viola	129-1740
Double Bass	48-240
Piccolo	548-3480
Clarinet	145-1381
Woman's Voice	193-870
Man's Voice	86-435

The foregoing table will give you an idea of the fundamental frequencies involved in producing sound. As will be pointed out in the next paragraph, these instruments also produce higher frequencies than those listed in the table. These are called harmonic frequencies or overtones. Thus, while a given sound may have a fundamental pitch of 240 cycles, it also includes other higher frequencies.

These other higher frequencies give to sound its identifying characteristics. This is a property of sound called quality or timbre. By means

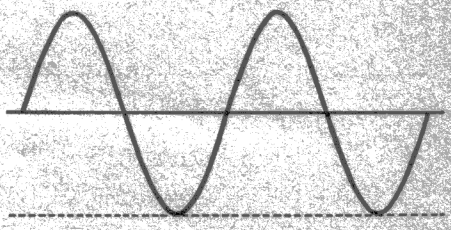
FIG.2

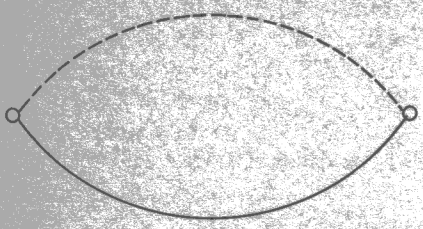

FIG. 3

of this property we can usually tell whether or not a voice is that of a man or a woman or whether a given musical note is produced by a violin, organ or piccolo. The same note or notes may be produced by all three of the instruments or voice, yet they do not sound alike. The reason this difference exists is because the different sources of sound in addition to producing the fundamental frequencies also produce varying amounts of other higher frequencies with varying strength or volume level. These extra *harmonic frequencies* determine the timbre or characteristic quality of all sounds which act on the human ear.

The fundamental frequency wave form shown in Fig. 2 is known as a *sine wave*, the name being derived from the particular mathematical method of developing the wave form. A musical or speech wave form is called a complex wave form because it is made up of the *fundamental frequency and several higher frequencies or harmonics.* A violin, flute, trumpet, harmonica, piano, etc., are all capable of producing the musical note of middle C which is 256 cycles. The actual sound waves from these instruments will not sound alike to our ears although they each contain a fundamental frequency of 256 cycles. The difference is the *number and relative strength of the harmonics or overtones* as previously mentioned. Now there is a definite mathematical relation between the fundamental frequency and its accompanying harmonics. If the fundamental is 256 cycles, twice this frequency will give the second harmonic. Three times the fundamental will give the value of the third harmonic, etc. Thus the second harmonic of 256 cycles is 2 x 256 = 512 cycles; 3 x 256 = 768 for the third harmonic; 4 x 256 = 1024 for the fourth harmonic; 5 x 256 = 1280 for the fifth harmonic, etc. Note in this the fundamental frequency is the first harmonic although it is usually called the fundamental.

The fundamental frequency as well as all of its harmonics have a basic sine wave form. However, all of these frequencies occur at the same time and our ears hear the sound as the *resultant wave form* of all the frequencies. This resultant is the equivalent of adding all the wave forms of a given sound to give one total wave form of sound to act on our ears.

It is important that you understand the mechanics of the production of harmonic frequencies. A study of Figs. 3 and 4 will help you to get the basic idea. These two figures represent a vibrating string such as that used on a banjo or simply a taut string stretched between two supports. In Fig. 3 is shown how the string might be vibrated or plucked at a very slow frequency so that, in effect, the string will swing over a given path in long slow loops. Even at a low frequency, the string in swinging back beyond its

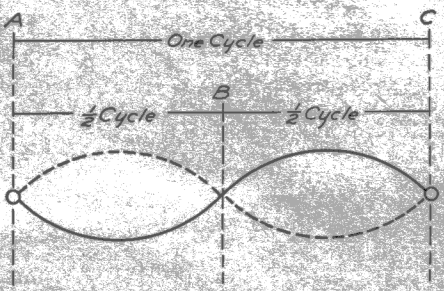

FIG. 4

original position is likely to have intermediate loops instead of the uniform loop in Fig. 3. Figure 4 shows the more likely condition. It is assumed here that the frequency is higher than in Fig. 3 so that in effect not only is the entire string vibrating throughout its length but different sections of it are also vibrating at other frequencies. For instance, from A to C in Fig. 4 may be considered the fundamental frequency of vibration. But note the two halves of the string as from A to B and from B to C are also creating half loops which result in the production of still another frequency of vibration. Thus a fundamental frequency from A to C is produced as well as the second harmonic from A to B and B to C. The A.C. equivalent of this type of wave form is shown in Fig. 5. Note the composition of the second harmonic in that it completes one full cycle (A to B) while the fundamental completes a half cycle in the same length of time. This proves the harmonic frequency is twice the frequency of the fundamental. The string analogy of Figs. 3 and 4 show simple illustrations of how a harmonic frequency might be produced. This, in itself, represents true conditions as far as it goes. However, musical notes and

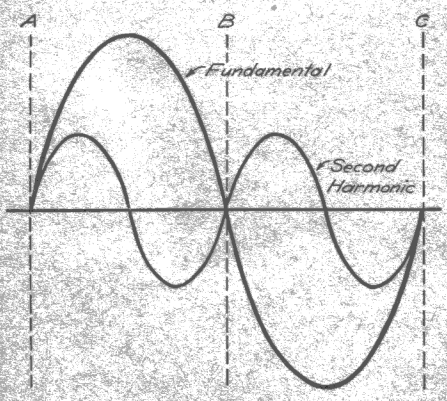

FIG. 5

speech are much more complex than Figs. 3 and 4 indicate. Whereas Fig. 4 shows how a piano string might vibrate to produce the second harmonic, actually the wave form it produces contains many more harmonics than just the second. The true conditions are when a piano string is struck there occurs throughout its length many separate vibrations (in theory every length of the string may vibrate) all of which blend together to give the characteristic sound of the piano. Thus, with different sizes of wire and different lengths of it, the whole musical range of the piano is made possible. The same principle holds true for other musical instruments and for the human voice. Columns or spaces of air are made to vibrate to produce sound. It should be clear, therefore, that the way the air is vibrated or set in motion determines the fundamental frequency and the number and relative strength of the harmonics. Another important principle of harmonics is that they cannot be fractional frequencies of the fundamental frequency. Thus, there is no such thing as a harmonic which is 1 1/2, 2 3/4 or 5 6/8 times the fundamental frequency. Harmonics are always even multiples of the fundamental frequency.

A distorted wave is one (it may be the resultant of several frequencies) which does not have a true sine wave shape such as Fig. 2. It may be odd shaped but as long as both its halves are symmetrical (both half cycles having the same general shape), it is not a distorted wave. *Distortion can be brought about in an amplifier by operating a tube on the wrong portion of its Eg-Ip curve, by overloading its grid circuit with a strong signal, by using improper or overloaded A.F. transformers or by otherwise operat-*

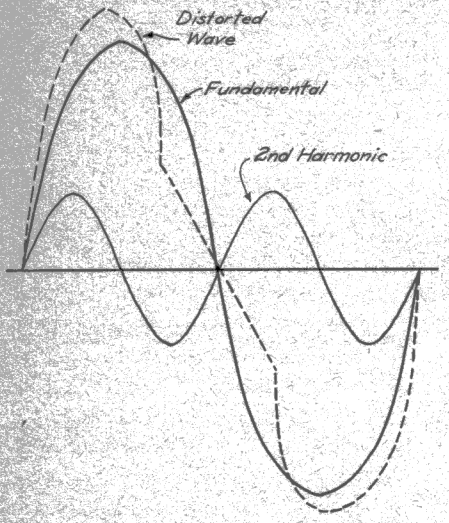

FIG. 6

ing the tube incorrectly. A tube so operated *will generate harmonics* of the fundamental frequency in its plate circuit and these will be reproduced by the speaker as distortion. Harmonics introduced into the grid circuit of a tube as part of the original signal do not constitute distortion, but harmonics introduced by the tube and its circuit will cause distortion in the sound output. The reason for this is that the resultant wave form of musical sounds or speech is, in general, symmetrical whereas the resultant wave form of harmonics, which is not a part of the original signal, is assymmetrical. A few examples will make this clear.

Figure 6 shows the resultant of an original sound wave, the fundamental frequency and its second harmonic which is in phase with the fundamental. Note particularly the dash curve. It is the result of adding the fundamental and the second harmonic. This resultant wave shape could very well represent a musical tone having a strong second harmonic.

Now a resultant wave of musical tones or speech may have a predominance of even (2nd, 4th, 6th, 8th, 10th, etc.) harmonics or the odd harmonics (3rd, 5th, 7th, 9th, 11th, etc.) may be predominant. Also, it is a fact that the whole combination of tones and frequencies making up a sound wave or its electrical equivalent contains the fundamental and probably both even and odd harmonics. In theory, there is no limit to the number of harmonics that may be present—some forty odd harmonics have been detected from complex sound waves. In practice, harmonics above the 4th are usually so small in amplitude that they need not be considered. Regardless of its complexity, a sound wave form can be resolved into an equivalent fundamental and a predominance of either even or odd harmonics. This is done mathematically by integral calculus. The *relative amount of energy in the harmonics* of the total wave form determines whether even or odd harmonics predominate.

To continue the study of resultant wave forms, refer to Fig. 7. This shows a fundamental, 2nd and 4th

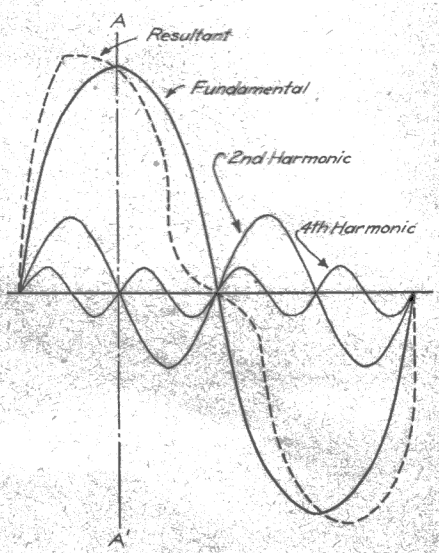

FIG. 7

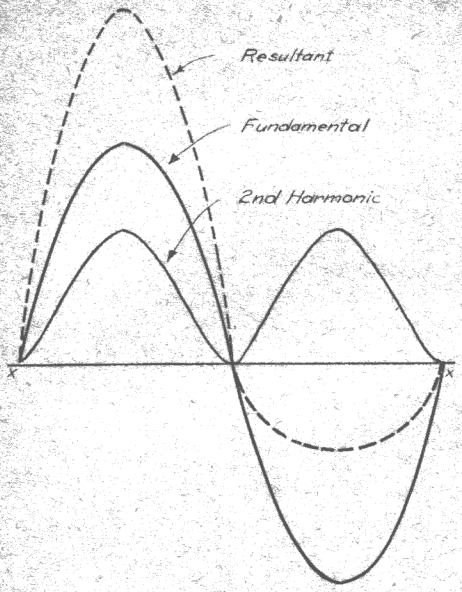

FIG.8

harmonic and a resultant wave form by dotted lines which, in turn, is the equivalent of the first three wave forms. This could easily be a piano or violin string musical note. Figures 6 and 7 are not necessarily distorted wave forms. While it is true that 1/4 cycles of each half of the resultant wave form are not alike, yet when complete half cycles of the resultant are balanced one against the other, the symmetry of the complete wave form begins to appear.

The next wave form of Fig. 8 does show distortion. This type of distortion results when a tube is worked on the lower portion of its Eg-Ip curve where *unequal halves* of the wave form appear in the plate circuit. See Fig. 9. This, in turn, means that a similar wave form would occur in an A.F. transformer whether it be used to couple to another tube or to a speaker. Note the strong second harmonic. *Later on we will show how the use of a push-pull transformer can be used to cancel out this second harmonic.*

Where odd harmonics are predominant, the resultant wave shape may be very similar to those produced by even harmonics. The wave form shown in Fig. 10 is one type of odd harmonic wave form. It shows the 3rd harmonic with its entire wave form above the zero or horizontal line. This is a rare type of wave form, but it may occur in some musical notes or combination of them. Figure 11 shows another type of wave form, containing a 3rd harmonic which is symmetrical about the zero line. In both Figs. 10 and 11 note the dash line is the result of adding the fundamental and 3rd harmonic to obtain the total result of the two wave forms. Still another type of musical tone wave form is shown in Fig. 12. This shows a fundamental, 2nd and 3rd harmonic plus the resultant of all three. Note the third harmonic is symmetrical about the zero line X-X¹. The second and third harmonics are out of phase in the first half cycle of the fundamental in Fig. 12, but they become in phase in the second half cycle of the fundamental resulting in the inverted dip of the resultant wave form. In this connection, A.F. phase difference is not detected by the ear but it does assume importance in television where sight is involved.

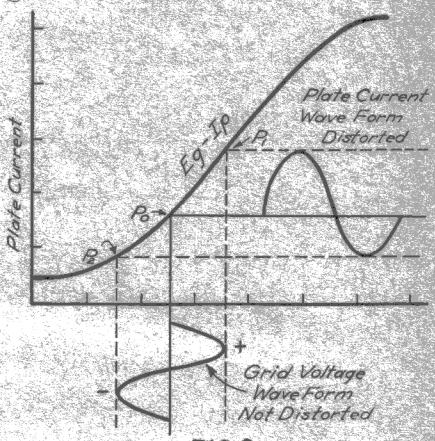

FIG.9

An important thing to remember about distortion and the wave forms of harmonics is that if the odd shaped wave forms represent the original signal *it is not distortion,* whereas if the tube or its circuits so alter the wave form to give it an odd harmonic shape, then it is distortion. This must remain necessarily true because all that amplification is supposed to do is to give an exact but magnified copy of the input at the output—any other wave form introduced into the signal wave form will be distortion and the sound from the speaker will be unnatural.

When a pure or perfect sine wave is distorted because of amplification, as described, the resulting wave, if analyzed, will be found to be composed of many other pure sine waves which may be considered separated from the fundamental wave. The sine wave is the basis of all wave motion—there does not exist any other kind of basic wave motion so far as is known. Some electrical equivalents of sound waves are so complex as to require an almost infinite number of sine waves to represent, but it is only in this way that they can be formed. *Due to the fact*

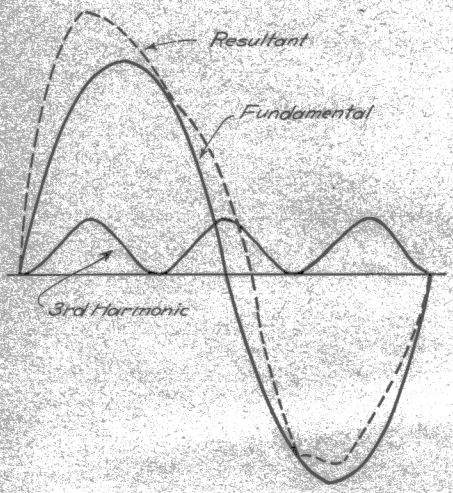

FIG. 10

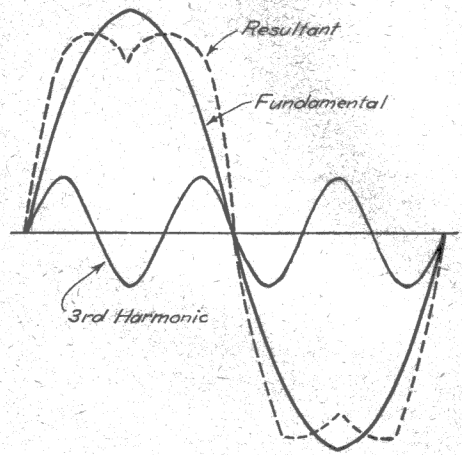

FIG. 11

that any kind of distortion will produce other sine waves (harmonics) always higher in frequency than the fundamental, the amount of distortion can be measured by the comparative energy in the harmonics. The ratio of the energy of the harmonics produced by distortion divided by the energy of the fundamental is the total distortion. This value multiplied by 100 will give the percent of distortion.

As you have already learned, the harmonics are exact multiples of the fundamental frequency—that is, 2, 3 and 4 times the fundamental, etc. Musically, the harmonics are harmonious. Since the 2nd harmonic is twice the fundamental (one octave) above it, the 3rd harmonic is three times the fundamental and the 4th harmonic (two octaves above the fundamental) is four times the fundamental frequency. The following table will show these relationships clearly.

Fundamental	
2nd Harmonic	1 Octave
3rd Harmonic	
4th Harmonic	2 Octaves
5th Harmonic	
6th Harmonic	
7th Harmonic	
8th Harmonic	3 Octaves

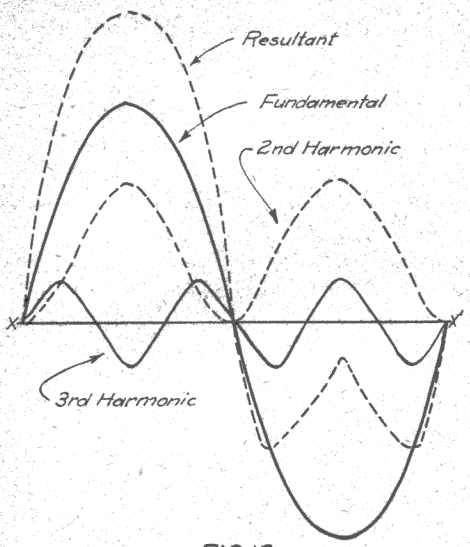

FIG. 12

Each harmonic is higher in frequency value by the amount of the fundamental than the next lower one and each time the fundamental is doubled an octave distance in the musical spectrum is covered.

Certain electrical circuits will produce a fairly large number of predominant even numbered harmonics such as 2, 4, 6, 8, 10, etc., and others will produce predominant odd harmonics: 3, 5, 7, 9, etc. Still other circuits produce both the even and odd harmonics. Generally, the lowest order harmonics, 2nd, 3rd, 4th, etc., *have the greatest amount of energy*. For example, if an amplifier produces 5% 2nd harmonic distortion, the other even harmonic percentages are likely to be 4th 2%, 6th .4%, 8th .08% and the others so low as to be quite negligible. The odd harmonics have about the same relative amount of energy and they all depend on each other for their relative amounts of energy. For example, if the 2nd harmonic energy is by some means reduced to 1/4th of its original value, it will also simultaneously reduce the 4th, 6th, 8th, etc., harmonics this much or more. Odd harmonics may be produced by tubes and their circuits, but are affected by the phase between the fundamental and even harmonic maximum peaks.

MEASUREMENT OF DISTORTION

To reduce distortion to a point where it cannot be detected by the ear of the average person, the total harmonic energy must be not more than 5% of the fundamental energy. By far the greatest portion of all of the harmonic distortion is contained in the 2nd and 3rd harmonics. If the 2nd is 4% and the 3rd is .8%, all of the others combined will probably have less than .2% and the total harmonic distortion will be less than the permissible 5%.

There are ways of calculating the harmonic distortion percentage based on values obtained from laboratory set-ups or from tube curves derived from these set-ups. The following formulas explain how the percentage of distortion is calculated. This holds true for either a single stage or push-pull stage of amplification.

(1) The percent of 2nd harmonic distortion equals:

$$\frac{(I_2 + I_1 - 2\ I_0) \times 100}{I_2 - I_1 + 1.41\ (.707\ I_2 - .707\ I_1)}$$

Where: I_2 = maximum plate current on positive grid peak.
I_1 = minimum plate current on negative grid peak.
I_0 = no signal plate current with normal grid bias voltage.

(2) The percent of 3rd harmonic distortion equals:

$$\frac{(I_2 - I_1 - 1.41)\ (.707\ I_2 - .707\ I_1) \times 100}{I_2 - I + 1.41\ (.707\ I_2 - .707\ I_1)}$$

(3) The percent of 2nd and 3rd harmonic distortion combined equals:

$$\sqrt{(\%\ 2nd\ H)^2 + (\%\ 3rd\ H)^2}$$

Where: The same values apply as explained in the foregoing.

In the first formula, the maximum and minimum plate currents are added and twice the average plate current or no-signal current is subtracted from it. This value is then divided by the difference between the maximum and minimum values of plate current plus the peak value of the difference between the R.M.S. maximum plate current and the R.M.S. minimum plate current. To obtain a percentage value from this factor it is multiplied by 100. The 3rd harmonic formula is somewhat like the one for the 2nd harmonic, and the total harmonic energy is the square root of the sum of the squares of the 2nd and 3rd harmonic percentages as expressed by formula No. 3.

These formulas are given mainly for reference. They are not considered very practical and certainly not very useful for the practical man.

The foregoing description explains the general nature of a musical, speech or noise wave form. With this knowledge as a background, it will help you to appreciate the fine qualities of push-pull amplification.

Before taking up the study of the push-pull amplifier you should first understand how harmonic distortion is generated by a tube and its associated circuit. Then with this knowledge it is easy to see the virtues of the push-pull system.

HARMONIC GENERATION

It has been pointed out that even the best single amplifier tube has slight curvature of its Eg-Ip characteristic, and because of this, a certain amount of distortion will be present in the output. In addition, if a strong signal drives the grid of the tube positive (which actually does occur in the 1st A.F. tube of many radio sets) distortion is created. Distortion may also result if the negative grid voltage of the tube is too low or too high. A.F. transformers can also cause distortion when their iron cores become saturated with plate current. This condition is reached when changes of plate current do not give equal changes of magnetism in the transformer core. Any of these causes of distortion will result in plate current changes which are not a true copy of the grid voltage changes. *Thus, the plate current wave forms take on shapes which are equivalent to the fundamental and undesirable harmonic frequencies.* From this condition we get the name of *harmonic distortion.*

One great virtue of the push-pull system is that it will cancel out all even number harmonics which are produced *as a result of the action of the tube or its circuit.* It is only the 2nd and 4th harmonics which have enough energy to give serious distortion trouble. Thus if these may be made to cancel as is done in the push-pull system, the total distortion is reduced to a minimum. You should understand that harmonics which are a part of the original signal are not cancelled by push-pull action. If this were done, the output wave form of a tube would not correspond to the input wave form. It is only the self generated undesirable tube harmonics which should be cancelled.

As already stated, there are several conditions that can occur in an A.F. stage to cause harmonic distortion. They all result in the creation of *extra unwanted harmonics* in the output wave form. Thus, we will consider the condition where *excessive grid bias* is applied to the grid of a

tube and let this serve to show how unwanted harmonics (regardless of their cause) are cancelled or balanced out altogether.

Figure 9 shows a graph of the Eg-Ip characteristics of a tube which has excessive bias. Note the operating point Po of the tube is forced to be low down on the curve where the curvature is excessive. When the grid swings positive due to a signal, *the grid, in effect, is made less negative so that* the corresponding half cycle plate current wave form is symmetrical with the grid voltage wave form because, in effect, the operating point for the peak of the signal is made to move to P1 on the Eg-Ip curve. When the grid swings negative it adds to the already existing excessive bias so that the operating point for the peak of the signal is forced to point P2 on the Eg-Ip curve. The corresponding plate current wave form for this half cycle as a result of the tube being worked on the curvature of its characteristic *has less amplitude* than the plate current wave form for the other or positive half cycle. As a result, the total wave form for the complete cycle is distorted or is not symmetrical. Confirm this for yourself by noting the plate current wave form in Fig. 9 has more amplitude on the top or first half cycle than it does on the lower or second half cycle.

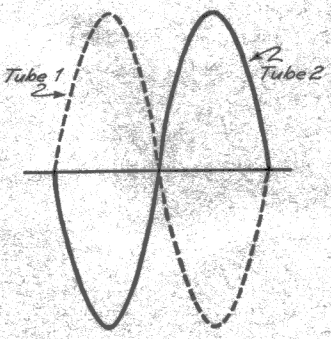

FIG. 13

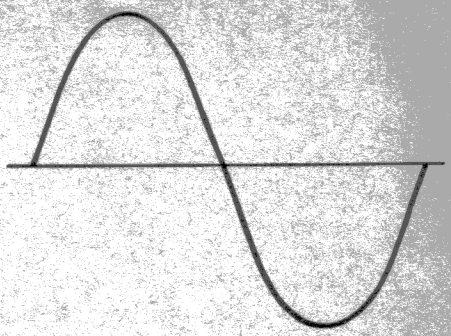

FIG. 14

When this condition occurs the tube is said to generate a strong second harmonic frequency. It is a long tedious process to prove this mathematically, but it can be done, and it can be demonstrated by means of an oscilloscope that such is actually the case. By way of explanation, an oscilloscope is a device similar to a television receiver which will show actual wave forms on its screen. You will learn more about this instrument later on in your S.A.R. course.

Later the operation of the push-pull circuit will be explained in detail. For the time being, just consider there are two tubes working together. They are so connected that when one tube has a plate current *increase* the other one has a plate current *decrease*. In other words, the plate currents of the two tubes are 180 degrees out of phase as shown in Fig. 13. It is assumed in this case that the plate current wave forms of the two tubes are symmetrical. Since the plate current wave forms are out of phase by 180 degrees they could be added to give an effective wave form of greater amplitude as shown in Fig. 14. The foregoing will serve to illustrate the principle of push-pull amplification.

To return to single tubes which generate second harmonic frequencies as a result of excessive bias, refer to Fig. 15. At A in this figure is shown

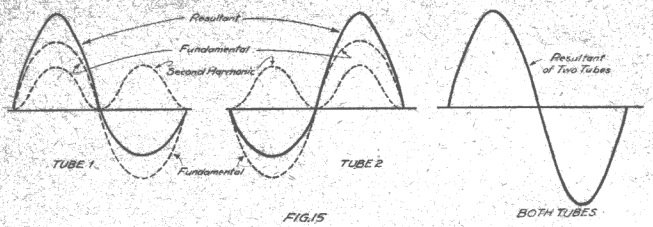

FIG.15

the plate current wave form of tube 1 of the push-pull system. Note there are three wave forms: 1, a fundamental; 2, a second harmonic; and 3, the resultant of the other two. From this you will see that the lower half of the fundamental is not symmetrical with the upper half. Next refer to wave forms B of Fig. 15. Note it is an exact duplicate of A except it is reverse in every way. It, too, has a fundamental, a second harmonic and a resultant of the other two. Both plate current wave forms are distorted, but remember the two distorted half wave forms *do not occur at the same time.* When tube 1 is creating its distorted half wave, tube 2 is creating its non-distorted half wave. When the grid voltage swings to the other polarity, the conditions are just reversed. As a consequence, if every point of one plate current wave form is added to every point on the other plate current wave form, the result will be a symmetrical wave form similar to C in Fig. 15 which will be practically free of distortion. It is in this way that all even numbered harmonics are cancelled out in a push-pull system regardless of what tube or circuit conditions cause them. Thus push-pull A.F. amplification is a definite improvement over single A.F. amplifier power tubes. This system does not eliminate third harmonic distortion, but it is usually negligible so that with push-pull the overall distortion is less than 5%. It is the second harmonic that causes most trouble because it has more energy as a rule than all the other harmonics combined.

PUSH-PULL AUDIO AMPLIFICATION

The next step in your study is the *push-pull amplification* circuit. This name has been given to a special type of circuit employing two tubes. The reason for its name is because the mechanical analogy of the electrical action of the circuit is very much like a mechanical device which pushes and pulls an object from a central point—first in one direction and then in the opposite direction. In an electrical sense, the signal is applied to the grid of first one tube and then the other as will be explained further on.

In Fig. 16 is shown a type 56 tube operating as a voltage amplifier feeding into two tubes known as a *push-pull circuit.* Transformer T1 is called the push-pull input transformer. It has a primary impedance Zp which is made as high as it is possible to make it and still maintain efficiency. The value of the primary impedance depends on the value of the load on the entire secondary which is small because it draws practically no current. The exact value of the secondary current is that required to charge the

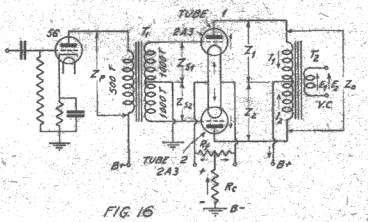

FIG. 16

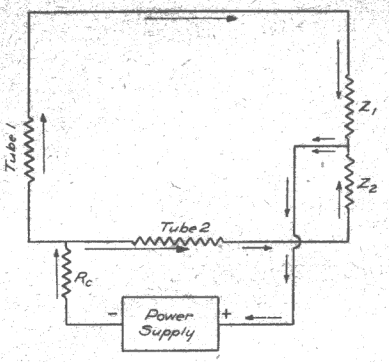

FIG. 17

small capacity existing between the grids and the filaments of the two tubes. For this reason the primary impedance (Zp) of T1 may be many times the A.C. plate resistance of the tube to which it is connected. You will remember from your study of voltage amplifiers that this will produce almost the full voltage amplifying power of the tube.

A definite analysis of Fig. 16 will now be given so that you may understand the basic principles of push-pull amplification. At ground where B minus is connected, electrons flow up from the A.C. power unit through Rc, as the small arrows indicate, to the centertap of a small hum balancing resistor Rf (20 to 50 ohms total). Next the current divides through each side of resistor Rf to the filament leads. From here electron flow is through both tubes from filament to plate and from each end of the primary of transformer T2 to its centertap where the electron path becomes common again and from there the path for current flow is back to B+ of the power unit supplying the circuit. The electrons or current divide into two identical parallel paths at Rf, each carrying substantially the same amount of current and then join again at B+. The electrical equivalent of this circuit is shown in Fig. 17 where resistors are used to represent the various circuit elements. Arrows show the direction of current flow from B minus to B plus. Study this circuit to get the direction of current flow firmly fixed in your mind. The two tubes in the push-pull stage are of the 2A3 type. Normally operated with minus 45 volts on their grids, the plate current through each tube will be 60 ma. *with no signal.* If each tube permits 60 ma. to flow, the total current from B— to B+ will be 120 ma. Flowing through Rc (bias resistor) this current flow will produce a voltage of 45 volts across it which is applied to the grids of the two tubes as a negative voltage. The value of Rc is equal to $\frac{E}{I}$ or $\frac{45}{.120} = 375$ ohms. If 295 volts is impressed between B+ and B—, there will be a 45 volt drop across Rc and a 250 volt drop across the two tubes from filament to plate. Since there is no D.C. current flow in the grid circuit, the grids are at the same voltage as ground, with no signal, hence are 45 volts more negative

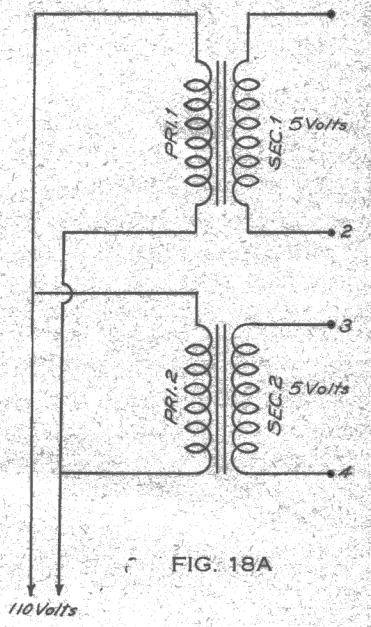

FIG. 18A

than the filament due to the voltage drop across resistor Rc. In other words, the grids are maintained at 45 volts negative with respect to the filaments (cathode in this case) just as other type amplifier tubes have their no signal steady negative grid voltage.

Now suppose that the primary of transformer T1 in Fig. 16 has 500 turns of wire and that the entire secondary has 2000 turns—that is, 1000 turns above the centertap which is grounded and 1000 turns below it. While the entire transformer has a ratio of 1 to 4 $\frac{2000}{500}$ *each grid of the 2A3 tubes receive only one-half of the total voltage produced across the entire secondary.* As far as it concerns the grids of the 2A3 tubes the transformer has a step-up ratio of 1 to 2 instead of 1 to 4. This is true of all push-pull transformers. Their ratios are considered as the ratio of the single winding to half of the centertapped winding, the reason of which will be brought out later.

It is important that you have a clear picture in your mind of the principles of push-pull *transformer action.* The figures of Fig. 18 show what takes place. At A is shown two step-down transformers which give a secondary voltage of 5 volts each. The primaries connect to the 110 volt line. If an A.C. voltmeter is connected across 1 and 2 or across 3 and 4 it will read 5 volts for each secondary. An important transformer principle is that the voltage of two secondaries will add to give a total voltage equal to the sum of the two if the phase of the two voltages is correct. This means both secondaries must be wound in the same direction. Thus, in Fig. 18A if we assume the two secondary windings are wound in the same direction, then connecting terminals 2 and 3 together will give a total of 10 volts across 1 and 4. In this case, the two voltages are in phase. Likewise, if, for instance, 1 is connected to 3 then across 2 and 4 the voltage will be zero because the two voltages are out of phase and will cancel one another. If it should happen that secondary 1 is wound opposite to secondary 2, then the connection of 2 to 3 will give zero volts across 1 and 4 and again there is cancellation. To get the voltages of the two secondaries to add with oppositely wound coils, 1 must connect to 3—10 volts will then appear across 2 and 4. In a push-pull transformer, you do not have to consider the direction of the windings, because the manufacturer arranges them correctly. The foregoing was included in order that you might be fully familiar with connecting transformer windings.

In Fig. 18A it was shown how two secondary windings were connected together to give a total equal to the sum of the two. In Fig. 18B a simplification of this circuit is shown wherein the two windings merely have one lead where the two windings join. Thus, between 1 and 2 there will be 5

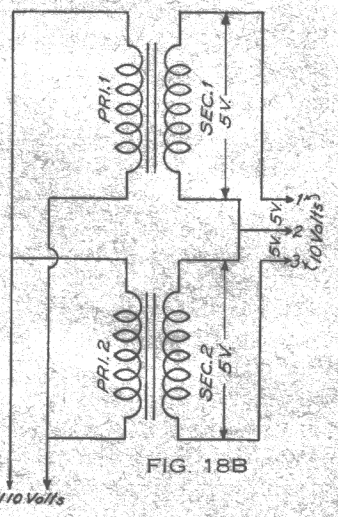

FIG. 18B

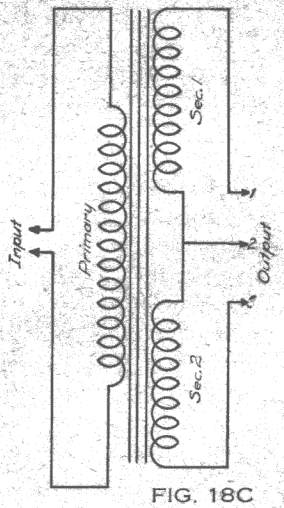

FIG. 18C

volts and between 2 and 3 another 5 volts is available. Across 1 and 3, of course, gives 10 volts. This connection is equivalent to connecting 2 and 3 together in Fig. 18A. Thus it should be clear that the two circuits give exactly the same results. Note in Fig. 18B terminal 2 is common to both 1 and 3.

So far, we have used two primaries, but this is needless because one primary winding properly arranged will induce just as much voltage in the two secondaries as will two primaries. Thus, the circuit may be further simplified to that of Fig. 18C wherein one primary is used. This circuit is still the equivalent of Figs. 18A and B. Since it is clear that two secondaries connected together give additive voltages, it follows that one winding with a wire tap at the center will do the same thing. So instead of going to the expense of constructing two windings, the manufacturer of the transformer merely makes one continuous winding and provides a centertap as in Fig. 18D. The centertap connection thus becomes common to each half of the entire winding. Thus, in effect, the transformer of Fig. 18D is one with a single primary and two secondaries. One secondary exists between 1 and 2 and the other one between 3 and 2.

The reverse condition also holds true wherein a tapped primary may be used as in Fig. 18E. In this figure, terminals 1 and 2 constitute one primary and 3 and 2 the other. Thus, current variations in either primary will induce a voltage in the single secondary because all of the windings are magnetically coupled. Note in Fig. 18E, terminal 2 is common to both 1 and 3.

The foregoing explains how tapped transformer windings may be arranged to give the effect of two separate transformers. Such transformers are used in push-pull circuits to give the desired effect. A push-pull transformer with a tapped primary as in Fig. 18E is called a push-pull *output transformer*. One with a tapped secondary is called a push-pull *input transformer* as shown in Fig. 18D. Thus in Fig. 16, T1 is a push-pull *input* transformer while T2 is a push-pull *output* transformer. With these principles of push-pull transformers clearly in mind you are ready to continue the story of push-pull amplification. Refer again to Fig. 16.

You have learned that if the grid bias on an amplifier tube is 45 volts

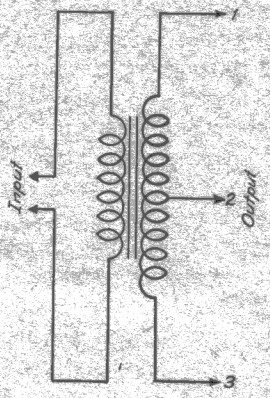

FIG. 18D

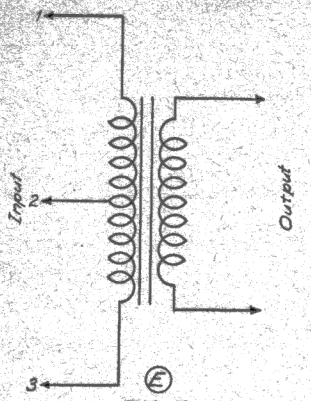

FIG. 18

negative, a *positive* signal voltage of 45 volts could be applied before the grid is reduced to zero voltage. Moreover, an additional 45 volts *negative* will charge the grid to a total of 90 volts negative. This is the *limit* of the grid voltage which can be applied to a 2A3 tube, as the grids must not become positive in this type of amplifier.

As the current in the primary of T1 of Fig. 16 is made to vary according to the changes of grid voltage of the 56 tube, a varying magnetic field is set up around the windings of T1. This causes a voltage to be induced in the entire secondary, and as the center point is grounded and cannot change its voltage, the induced voltage is distributed with equal and opposite values to the grids of the 2A3 tubes. Remember this must be true, for a coil with an induced voltage cannot have both of its ends of the same polarity at the same time. If one end is positive, the other must necessarily be negative and vice versa. When the maximum of one alternation is reached, the grid of the top 2A3 tube (No. 1 of Fig. 16) will be 45 volts more positive or at zero volts and the grid of tube No. 2 will, at the same time, be 45 volts more negative or minus 90 volts. In the first tube, the signal voltage of 45 volts has subtracted from the D.C. bias voltage, and in the second tube it has added to the bias. The entire transformer secondary has produced a total of 90 volts from end to end with the lower end negative with respect to the upper end. At the center, it is 45 volts more positive than at the lower end.

Now what happens to the plate current of the two 2A3 tubes when the grid voltage changes as described? When the grid of tube No. 1 is brought to a zero voltage, the plate current of it *increases to about 110 ma.* Since at the same time the grid of tube No. 2 has a charge of —90 volts, the plate current of it *reduces to about 10 ma.* You should understand that when the two primary currents I_1 and I_2 in transformer T2 of Fig. 16 are equal or both at 60 ma., *there is no total magnetism in the transformer core.* Equal currents are flowing in the two halves of the primary of T2 in opposite directions—*hence cancelling all magnetism.*

As you have just observed, a signal applied to the grids of the two tubes will *increase one of the currents and decrease the other.* In this case, the state of *magnetic balance* no longer exists, as one current is much greater than the other. Furthermore, you know that a voltage is induced when magnetism is changing and not when it is at any fixed value.

The increase in current of I_1, from 60 to 110 ma. induces a voltage E_1 in the secondary. A simultaneous decrease in I_2 from 110 to 60 ma. (assuming a continuously applied signal) induces a voltage E_2 in the secondary *which is in the same direction as* E_1. Although the currents are flowing in the opposite directions in the primary of T1, you will recall that an increase of magnetism produces a voltage in one direction while a decrease of current in the same direction induces a

voltage in the opposite direction. Therefore, when the *currents* are in opposite directions an increase in one will induce a voltage of the same polarity as a decrease in the other. The two voltages, therefore, add to or reinforce one another just as was explained for Fig. 18A. In addition to this, there is also the effect of mutual inductance which still further increases the net voltage output of the secondary T2 in Fig. 16.

During 11 selected points of *one A.C. alternation* the current will vary through the two push-pull tubes *approximately* as indicated by the following table:

	Current in Tube No. 1	*Current in Tube No. 2*	*Total Current*
1	60 ma.	60	120 ma.
2	70 "	50	120 "
3	80 "	40	120 "
4	90 "	30	120 "
5	100 "	20	120 "
6	110 "	10	120 "
7	100 "	20	120 "
8	90 "	30	120 "
9	80 "	40	120 "
10	70 "	50	120 "
11	60 "	60	120 "

It is quite obvious from the foregoing table that the *sum of the two currents* in the parallel paths of the transformer (see Fig. 17) is *constant*. This is actually true within about one percent. The amount of current drawn from the power supply is essentially constant and the current through Rc in Fig. 16 is supposed to be uniform at all times. No by-pass condenser is needed across the bias resistor for this reason. If the Mu of one tube changes, or if one tube gets older or weaker than the other in a push-pull stage, there will, however, be a slight ripple voltage across the bias resistor Rc.

During the next A.C. alternation, tube No. 2 acts exactly as did tube No. 1 and, in turn, tube No. 1 acts as did tube No. 2. The two 2A3 tubes generate A.C. signal power from the D.C. plate current of the two tubes which is drawn out of the transformer through the secondary V.C., called the *voice coil* winding supplying the speaker with power.

As you know, there are three common ways to calculate the power in a circuit, namely, $P = E1$; $P = I^2R$ and $P = \frac{E^2}{R}$. Some of these values, such as I, in many cases can only be estimated, but they all will yield the same results if correct values are used.

For example, in finding E across the primary of T2 for the first formula, you must find the effective gain of one tube—multiply this by the grid voltage and double the result because the voltage of one plate reduces as much as the other increases, doubling the total voltage difference from plate to plate. This multiplied by the net plate to plate current will yield the power output in watts.

Although using a load resistance equal to the tube's A.C. plate resistance will provide maximum power transfer to the load, this can rarely be done because of the distortion inherent in the operation of the tube. This is due to the curvature of the Eg-Ip curve of the tube as already explained.

As the load is increased above that of the Rp of the tube, the plate current becomes more and more proportional to the grid voltage and the distortion is reduced. For this reason the load value is chosen by actual experiment which is a compromise between wanted power and percent of distortion. For a 2A3 tube, the A.C. plate resistance Rp is only 800 ohms, but a load of 2500 is used so that the

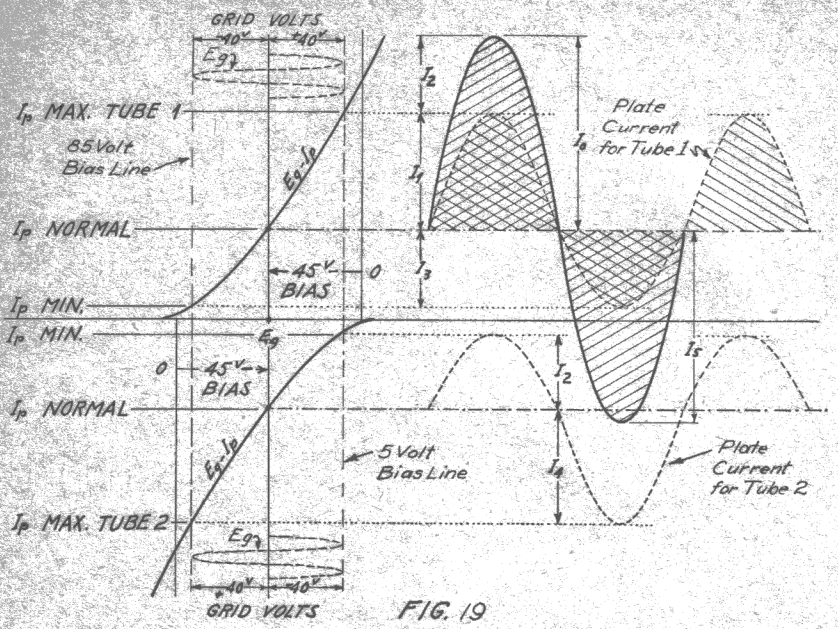

FIG. 19

distortion may be reduced to an insignificant minimum.

With this information, the actual amplified voltage for the push-pull stage under consideration will be

$$\frac{Mu\ Eg\ RL}{Rp+RL}$$

or

$$\frac{4.2\times45\times2500}{800+2500}=143.2$$

volts for one plate or 286.4 from plate to plate of which the R.M.S. value is 202.5 volts from plate to plate. The peak current will be 110—10 or 100 ma., which must be considered as flowing in only one-half of the coils as this is what actually occurs. The R.M.S. current is 100 × .707 or 70.7 ma., and one-half of this through the entire coil is the same as all of it flowing through one-half of the coil. Therefore, $\frac{70.7}{2}$ = 35.35 ma. is the current which must be considered with a voltage of 202.5 acting from plate to plate. 202.5 × .03535 = 7.15 watts approximately as the power output of the two tubes.

You can find the power more exactly from $P=\frac{E^2}{R}$ — where E = 202.5 and R = 5,000 ohms for the complete plate to plate impedance. 202.5^2 = 41,100 and $\frac{41,100}{5000}$ = 8.22 watts. This is more accurate than the other computation where the exact current in the primary is not known.

Now transformer T1 in Fig. 16 is a 2 to 1 step-up transformer requiring only 22 1/2 volts across its primary to produce a 45 volt grid swing for the 2A3 tubes and hence 8.22 watts output. If the primary impedance Zp is 8 times the Rp value of the 56 tube, the voltage amplifying ability of the 56 tube stage should be approximately .9 times the amplification factor of the 56 tube which is 13.8. The product of .9 and 13.8 gives 12.4 and this divided into 22.5 volts gives an answer of 1.813 volts required at the 56 tube

grid to produce 8.22 watts power output across the secondary V.C. of the output transformer.

For practical purposes the power output may be regarded as 20% of the plate input or 1/5th of the product of plate voltage and plate current. The ability of the triode to convert D.C. power into A.C. power represents an average efficiency of 20%. With this calculation the power output would be equal to:

$$\frac{250 \times (.060 + .060)}{5} = 6 \text{ watts}$$

You are now ready to study how a push-pull amplifier automatically eliminates some of the distortion which is inherent in a single tube stage. Two Eg-Ip curves have been drawn in such a way in Fig. 19 that the operation of both tubes can be observed simultaneously. The two —45 volt bias lines have been brought together and these cross the two Eg-Ip curves at the proper operating point. When one signal alternation is impressed on the grids they are both driven, in Fig. 19 for instance, to the left, one tube has an increasing negative grid voltage while the other tube has a decreasing negative grid voltage. The reverse, of course, holds true when the signal alternation changes polarity. Note that the lower Eg-Ip curve is reversed from the upper one for purposes of illustration, and *this makes it appear that both grids are acting in the same direction*, but actually, of course, this is not true. Now that you understand how the grid voltage wave form in Fig. 19 acts, suppose a signal of 40 volts is assumed for the Eg wave form. Next consider that this will, at first, drive both grids to the right in Fig. 19. In tube No. 1, the bias will be reduced to 5 volts when the signal reaches a peak value of 40 volts, and the plate current will rise to I_1 due to the shape of the Eg-Ip curve. The grid of tube No. 2 on the the same alternation will be driven 85 volts negative —45 + (—40) = —85, and the plate current of this tube will drop to a value represented by I_2. Note carefully that this decrease of I_2 is somewhat less than the increase of I_1. As explained, because the two currents are in phase, I_2 will add to I_1, producing a total equal to I_0. The addition of I_2 to I_1 is clearly shown by the shaded wave forms of tube 1 in Fig. 19. Note carefully how $I_1 + I_2 = I_0$ with respect to the amplitudes of the current wave forms of tube 1.

Now on the next alternation, tube No. 1 produces the smaller variation and tube No. 2 the larger one. This time, the grids swing to the left, making the plate current of tube No. 1 drop to I_3, but making it rise to I_4 in tube No. 2. The larger value I_4 is added to the smaller value I_3 in this case producing I_5 which has exactly the same amplitude as I_0, but in the reverse direction. You will note each tube suppresses the current wave form on its most negative grid swing. The other tube, however, comes to the rescue of the one having the distorted half cycle and fills it in to the correct amplitude. It is in this way that one tube helps the other and, as a consequence, both half cycles are symmetrical and second harmonic distortion is eliminated.

A typical *pentode* push-pull power amplifier circuit is shown in Fig. 20. Its operation is substantially the same as that of the triode amplifier just studied. The screen grids are connected directly to B+ and the suppressor grids are connected directly to the cathodes.

The voltage gain of the push-pull section is somewhat greater than the triodes requiring less input to the control grids for a given power output.

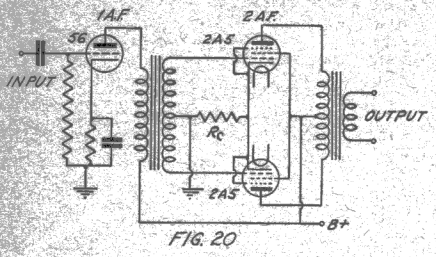

FIG. 20

Usually the voltage output of the detector which precedes the audio amplifier has sufficient voltage to feed right into the push-pull pentode output stage. Thus some receivers have only one audio amplifier stage. There are very rarely more than two stages in modern receivers.

In Fig. 21 is shown one practical method of coupling a resistance-capacity coupled stage to a push-pull power amplifier. It should be apparent to you at this time that there are certain advantages in using resistance coupled amplifiers. These advantages are partly preserved in Fig. 21. The signal voltage developed from plate to ground of tube P is impressed through the coupling condenser C through the upper half of the choke (Ch) to its centertap (Ct) which is grounded.

Thus an equal and opposite voltage is developed from Ct. to the other grid. With this arrangement the circuit of Fig. 21 becomes the equivalent of Figs. 16 and 20.

Not all push-pull circuits require transformers. An arrangement of this type is shown in Fig. 22. Here the

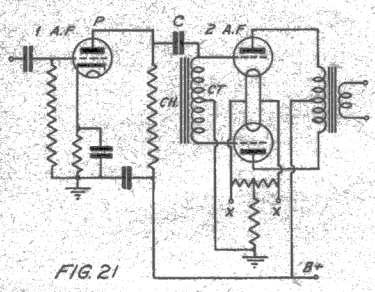

FIG. 21

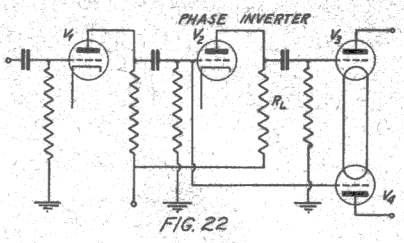

FIG. 22

first audio amplifier V1 feeds the amplified signal to the grid of V2 as usual and also to the grid of tube V4. The same attenuation (meaning either an increase or decrease in the signal) which makes the grid of V2 positive will naturally make the grid of V4 positive, but the grid of V3 will be negative, because as the grid of an amplifier goes more positive in potential, the plate becomes more negative. Thus the proper phased grid signals are fed to both of the push-pull tubes. The gain of the tube V2 must be reduced to 1 so that the same amount of voltage may be fed to V3 as to V4. This is done by making resistor RL in the V2 plate circuit low or by only using part of its as a load.

The extra tube V2 used in this circuit reverses the phase of the signal for V3 and is, therefore, called a *phase inverter* tube. Phase inversion comes about due to the fact that the signal in the plate circuit of a tube is 180 degrees out of phase with the signal fed to the grid circuit. Thus, if the grid of tube V4 in Fig. 22 is connected to the same point as the

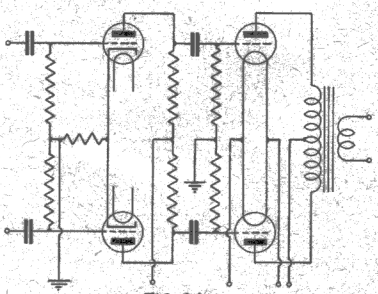
FIG. 23

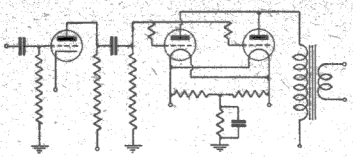

FIG. 24

grid of V2, one phase of the voltage is established for V4. Then if the output of tube V2 is made to feed the grid of V3 the phase of the voltage fed to V3 will be correct because V2 reverses the phase of the signal voltage just as any amplifier will do. Thus, correct phase is established for both tubes of the push-pull system consisting of tubes V3 and V4.

In Fig. 23 is shown a push-pull resistance coupled amplifier stage of the push-pull type coupled to a push-pull output power system. While triode tubes are shown in Fig. 23 pentode tubes may also be used as in Fig. 20 with resistance grid coupling. Two stages of push-pull amplification are rarely used in radio sets, but you will find them in use in many public address systems and also in other small A.F. amplifiers.

In order to increase the power output of an amplifier without using a push-pull circuit, tubes are used in parallel occasionally. An output circuit using two tubes in parallel is shown in Fig. 24.

When this system is employed small resistors are placed in series with each control grid, ranging from 20 to 500 ohms to prevent the action of one tube from affecting the other. Small amounts of secondary grid emissions have a vital effect on the tendency toward grid current flow. With these resistors the two tubes, because of slight differences in characteristics or operating voltages, may be somewhat different in this detail without affecting each other.

Still another method of obtaining considerable power output is shown in Fig. 25 where four tubes are used in parallel-push-pull. While triodes are shown in Fig. 25, you should remember that pentode tubes may be wired in this way as well as triodes.

A.F. CIRCUIT REFINEMENTS

Due to advancement in the knowledge of A.F. circuits several refinements are in common use. Figure 26 shows some of these. At A you will note there is a resistance R in series with C carrying the signal. Its value may be anywhere from 10,000 to 250,000 ohms. Although it cuts down the signal to some extent, the total impedance of R and Xc will be fairly constant over the entire audio spectrum. This is because R is very large as compared to Xc and large changes in the latter because of frequency changes will affect the total A.F. range a minimum. In this way all frequencies will be amplified very nearly the same amount.

At B is shown a typical and widely used *plate circuit filter*. RL is the tube load and R and C form a filter so that the extra current required for changes in plate current may be drawn instantly from C without changing the voltage or current at B+ in the least. R is usually somewhat smaller than RL, ranging from 10,000 to 100,000 ohms. Grid filters are commonly designed in the same way to prevent signal currents in the grid resistor from affecting the D.C. bias and to prevent bias ripple voltage from affecting the grid. See Fig. 26C.

A similar grid filter for transformer

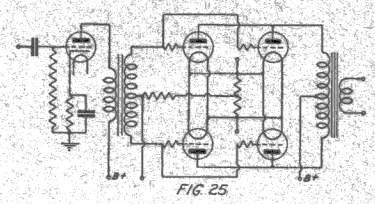

FIG. 25

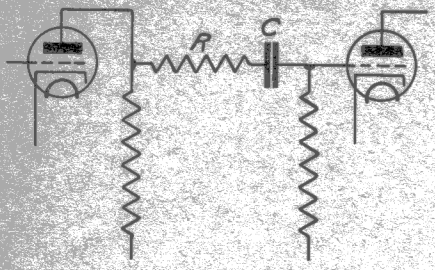

FIG. 26 (A)

or choke coupling is often used as in D of Fig 26. The value of C for circuits of Fig. 26B, C, or D may be anywhere from .01 mfd. to .5 mfd. This type of filter is also commonly used in the plate circuit of many amplifiers. Volume control is usually desired in the audio amplifier. This is accomplished through the use of a potentiometer as at C of Fig. 26. A portion of the signal voltage across R is selected for the grid, depending on the portion of resistance used through the setting of the sliding contact.

Circuits E and F of Fig. 26 show the variable and fixed methods of *tone control.* In circuit D the shunting effect of the combination of R and C determine the portion of energy at every frequency which is short circuited and hence does not get to the

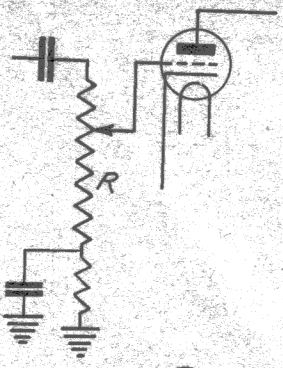

FIG. 26 (C)

speaker. As the amount of R in the circuit is reduced, the shunting effect advances rapidly with frequency but as R is made much larger, the shunting effect is more nearly uniform for all frequencies.

The tendency for amplifiers to favor high frequencies in general is compensated for through the use of condenser C as at F of Fig. 26. It shorts out successively more and more energy from higher and higher frequencies as the value of Xc reduces with frequency increases. This compensates for the tendency of the amplifier to give the higher A.F. frequencies more amplification than the lower frequencies.

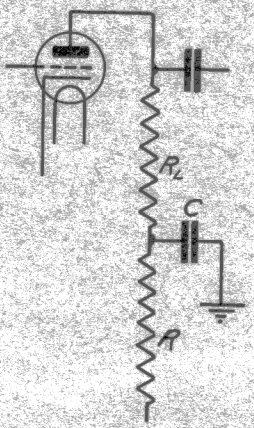

FIG. 26 (B)

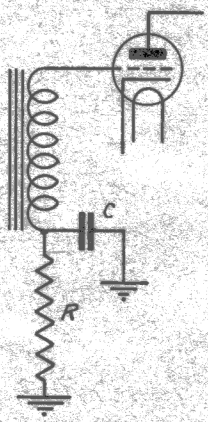

FIG. 26 (D)

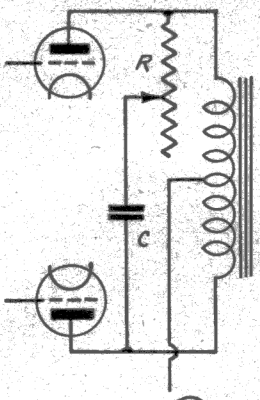

FIG. 26 (E)

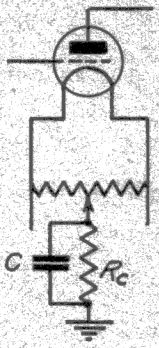

FIG. 26 (G)

At G of Fig. 26 is shown a filament potentiometer with a sliding contact for hum adjustment. Quite frequently any plate or grid ripple voltage which may get into the signal circuit may be reduced considerably or eliminated completely by *adding* a small ripple from the filament supply. The slider of this adjustable resistor is simply moved to the position of minimum hum when an adjustment is necessary. The purpose of Fig. 26H is similar to that of Fig. 26F in that the condenser C shorts out more and more energy of the higher A.F. frequencies as the frequency increases. In essence, it is a filter for the higher A.F. frequencies.

In the circuit of Fig. 26I, the bias resistor and grid filters are combined. Rc serves to bias the tube, while R and C together act as the grid filter system.

The circuit of Fig. 26J is similar to Fig. 26F and H, but in this case the resistor R is adjustable to provide for tone adjustments most pleasing to the individual ear. Later lessons in your S.A.R. course will give more details on tone controls.

Finally, in circuit K of Fig. 26 there is shown a special circuit for high quality audio amplifiers. The input transformer secondary is split into two separate windings so that each tube may be biased separately from an external source. Exact amplifier balance can be maintained in this way. A milliammeter connected in series with each plate circuit will show when the correct balance is obtained.

AMPLIFIER CLASSES

During your studies of the push-pull system or of amplifiers in general it has undoubtedly occurred to

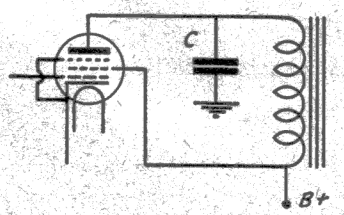

FIG. 26 (F)

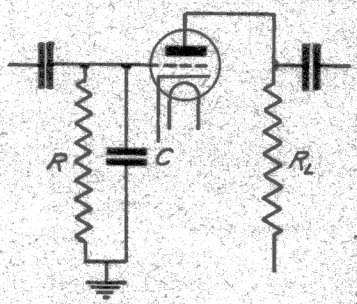

FIG. 26 (H)

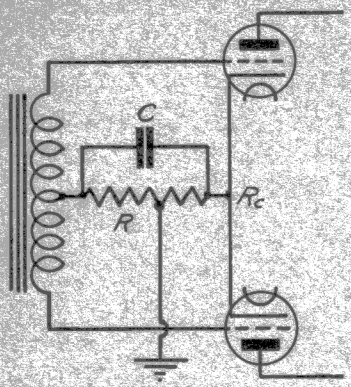

FIG. 26 (I)

you that all such amplifiers are of fairly low efficiency. For example, only 20% of the power supplied to the plate circuits of a triode push-pull circuit could be converted into A.C. signal power. A good portion of the available grid voltage (within its power to accurately control the plate current of the tube) is not used. Not only is there the complete range on the Eg-Ip curve from zero grid to plate current cut-off, but also there is a large range available in the positive grid region of the tube provided it can be effectively used. Figure 27 shows a method in which the total control range of the grid (from maximum negative to maximum positive) may be used.

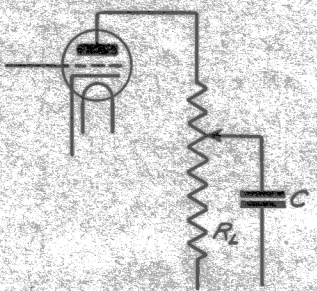

FIG. 26 (J)

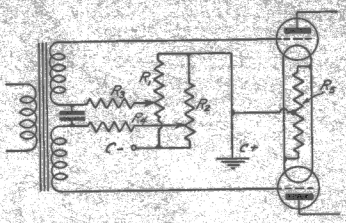

FIG. 26 (K)

Here again, as in Fig. 19, the Eg-Ip curves of the two tubes have been placed so that the no-signal grid voltages of each tube of the push-pull system coincide. This time it will be noted that both tubes are biased almost to their cut-off points indicated by the vertical line G_o. Now when the grid voltages advance to the right from G_o to G_1, for instance to 80 volts, the following actions take place.

When the grids have advanced up to about 20 volts (on their way to a maximum of 80 volts) in this case with a normal bias of —50 volts, the plate current of tube No. 2 is completely cut off. As the grid continues in the positive direction for tube No. 1, its plate current *increases* steadily. When the zero voltage grid value is reached for tube No. 1, which corresponds to about —165 volts for tube No. 2, grid current flows in tube No. 1 as shown by the dotted line, and tube No. 2 remains *inoperative*. The *grids* in this type of amplifier are supplied with *power* from the preceding stage so that their voltage will remain as intended even while grid current flows. The grids continue No. 1 in the positive direction, No. 2 in the negative, until the former reaches +130 volts and the latter —300 volts. In spite of the grid current, there is still adequate plate current and it rises to a maximum of 300 ma. as shown. Reducing to zero again, the grid signal voltage has completed one alternation.

The second alternation of the grid to the left of G_o to G_2 involves the

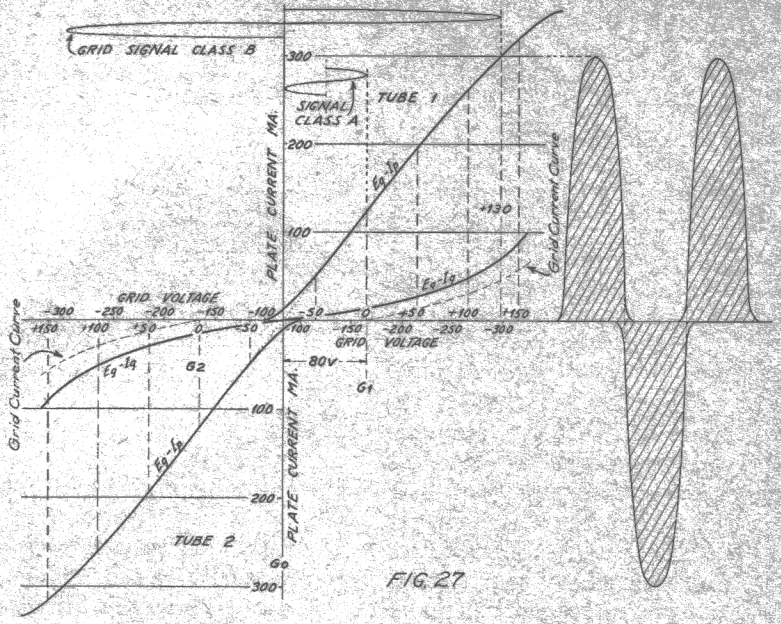

FIG. 27

operation of tube No. 2, just as tube No. 1 operated during the first alternation. The first tube handles all of the signal of one alternation and the second one handles all of the signal of the opposite alternation.

To distinguish this class of amplification from that which you have been studying up to now, it is called *class B amplification;* the other, such as that shown in Fig. 16, being named *class A* as it was the first type used. Just compare the grid voltage swing of a class B amplifier to that of a class A amplifier—the wave forms of both being shown in Fig. 27 for direct comparison. The grid voltage swing of a class B tube is often 4 or 5 times as large as the class A and uses virtually all of the range of control of which the grid of a tube is capable of handling.

In Fig. 28 is shown a typical class B circuit using an ordinary triode tube. While it is not unlike an ordinary class A circuit, there are several important differences. It was mentioned that the grids must be supplied with power as they draw current and that their voltages must be maintained. Thus the grids may not be resistance coupled to the preceding amplifier, but must make use of a power amplifier with *step-down transformer* and not just a voltage amplifier. Because it serves to *drive* the push-pull grid voltage positive, the first stage is called a driver stage. The driver stage may be a transformer coupled class A, single or push-pull power amplifier using triodes or pentodes. The impedance used in the driver plate circuit must be reduced for the class B grids and a step-down transformer

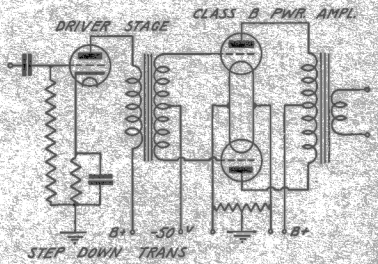

FIG. 28

is invariably used because the amplified signal voltage must be translated into power and only a transformer may do that.

Referring again to Fig. 27, it is evident that the plate current flowing when no signal is being supplied to the grids is very low—in fact, it is practically zero. When a maximum signal is applied, having a peak value of 130 volts, the sum of the currents of the two tubes having a peak of 300 ma. will be 300 × .707 or 212 ma. Remember that only one tube is working at a time, and hence the two currents cannot be added. The amount of current drawn from the power supply depends directly on the signal intensity which it handles. Thus a class B amplifier places a heavy current demand on its power supply when the grid voltages are large.

In a class A amplifier you have learned that the average current through both of the tubes was practically constant for any signal within its operating range. On the other hand, another condition is present with a class B amplifier requiring a good power supply system which is capable of supplying various amounts of current over a wide range instantaneously without appreciable voltage changes. Thus, a mercury rectifier tube is often used for class B systems.

Obviously, a self-biasing resistor cannot be used in the filament center-tap for a class B circuit, because the plate current varies so widely that the bias would change from zero to maximum during every half cycle of the grid signal. For this reason, a separate bias supply which does not depend on the plate current of the class B circuit must be used.

The difficulties encountered in connection with class A circuits led to the development of tubes for class B operation. The first of the series was the 46 tube, which is an output tetrode. The 49 is the same type in the 2 volt filament class. Although this tube has two grids, a filament and a plate, it is always operated as a triode. For class A operation, the outer grid is externally connected directly to the plate, but for class B operation, the outer grid is connected to the inner or control grid.

The characteristics of a 49 tube as a class A amplifier are very similar to other power triodes, but as a class B amplifier where two tubes must be used, they are very different. With no grid bias supplied and 400 volts on its plate, the plate current will be only 6 ma. The two grids are designed to practically cut off plate current with no bias. This is a matter of internal construction of the elements. The tube was designed in this way so that its use would not be limited to class B operation only.

A class B circuit having an output of 20 watts is shown in Fig. 29, using two type 46 tubes in a class B power amplifier circuit with a type 46 tube used as a class A driver. The following details are to be noted in particular. The driver may be resistance coupled with respect to its input as it is a class A amplifier of the power type drawing no grid circuit.

Its *average* plate current is constant and may be self-biased with Rc as shown. This resistor is by-passed with condenser C so that the bias variation

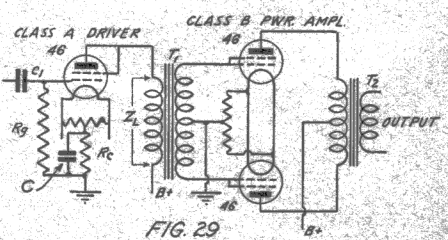

FIG. 29

is at minimum. The load ZL on the driver is about three times the A.C. plate resistance of the tube when wired as a class A amplifier for maximum undistorted output. The outer grid is connected to the plate for this class of operation. The output of this driver is normally 1.25 watts. The class B stage input coupling transformer T1 is a step-down type and must supply a minimum of .65 watt to the class B grids for 20 watts output. Here the two grids of each tube are connected together, both acting as a single control grid. Both the filament centertap and the grid return are directly grounded since no bias is required for the grids.

For a maximum signal the average plate current will be 140 ma., while it will be only 12 ma. for both tubes with no signal. The current will reach peaks of 200 ma. in one tube at full power output. The plate to plate load impedance furnished by the output transformer T2 is only a small percent higher than the plate to plate A.C. plate resistance of the tubes in this circuit for maximum power output with minimum distortion.

The shape of the plate current curves in Fig. 27 is due to the gradually tapering *cut-off characteristics* of these circuits, and indicates that distortion will be produced. Rather than being proportional to the signal level, however, the smaller the signal the greater will be the percentage of this distortion to the entire output. Therefore, the distortion decreases with increased output due to this effect. Other sources of distortion, principally due to the bending of the upper portion of the Eg-Ip curve have an increasing effect, however, on distortion at high volume levels. The class B amplifier has, in general, somewhat more inherent distortion in practice than the class A type.

Because all class B circuits required the use of two tubes, it soon became common practice to produce *single tubes for class B operation with two sets of twin elements in them.* These were made for filament voltage operation. They are the 19 tube for 2 volt operation; the 53 in the 2.5 volt A.C. class and the 6A6 or 79 and 6N7 (metal tube) in the 6.3 volt class.

There were also tubes developed with three grids to be operated as class A triodes, as class B triodes or as class A pentodes. For class A triode operation, the two outer grids were connected to the plate. For pentode operation, the three grids were used in the regular pentode manner with the inner grid for signal control, the second grid for a screen and the third as a suppressor, connected to the cathode. As a class B triode, two tubes are used with the inner and middle grids connected for control and with the outer grid connected to the plate. Only two tubes of this type were developed, one for the 2.5 volt A.C. filament and one for the 6.3 volt class. These are the 59 and 89 respectively.

It must be remembered that an ordinary triode originally used for class A work can be adapted to class B operation and that the curves of Fig. 27 cover all types. Where a class B type of tube is used the line G_o represents the zero bias line, whereas for others it represents the negative bias which must be added for plate current cut off. To find the bias required for any tube used as a class B tube use the following formula. Bias voltage (class B) equals zero grid-plate current times class A bias voltage divided by zero grid-plate current minus class A plate current. As an

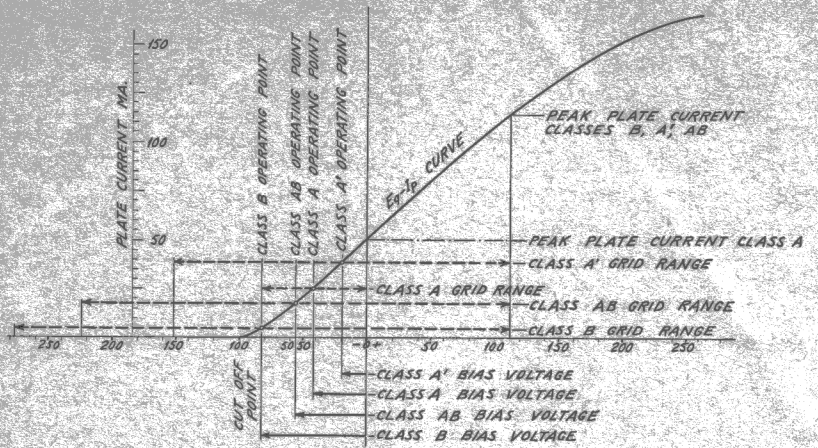

FIG. 30

example for a 2A3 tube, this would be:

$$\frac{.12 \times 45}{.12 - .060} = \frac{5.4}{.06} = 90 \text{ volts.}$$

There is still another kind of amplifier known as a class A prime, written also A′, which uses a lower bias than the class A type tube and hence is allowed to cover more of the negative region of the grid control range of the tube. On the positive swing of the signal, the grid becomes positive and must be supplied with power as for class B operation. Two tubes must be used with this type of amplifier. It has less distortion than the class B type, but is less efficient.

There is also the class AB amplifier which derives its name from the fact that its bias lies somewhere between the class A and class B values. Fig. 30 shows a typical Eg-Ip characteristic curve with the various operating points and the maximum grid signal which can be supplied for each class of operation. The peak plate current for each type and the relative bias values are clearly indicated. Study Fig. 30 in careful detail until you are sure you understand the operating principles of the various classes of amplification.

These questions are designed to test your knowledge of this lesson. Read them over first to see if you can answer them. If you feel confident that you can, then write out your answers, numbering them to correspond to the questions. If you are not confident that you can answer the questions, re-study the lesson one or more times before writing out your answers. Be sure to answer every question, for if you fail to answer a question, it will reduce your grade on this lesson. When all questions have been answered, mail them to us for grading.

QUESTIONS

No. 1. Why will a push-pull amplifier produce less distortion than a single stage of amplification?

No. 2. How may a single stage amplifier be coupled to the input of a push-pull stage without using a transformer or choke coil?

No. 3. What is a complex wave?

No. 4. What is the maximum percentage of total harmonic distortion that may be tolerated by the human ear?

No. 5. When tubes are wired in parallel what may be done to equalize their relative amplifying ability?

No. 6. Why is not the self-bias arrangement employed for class B amplifier tubes?

No. 7. Compare briefly, class A and B amplifier circuits as to bias and power output.

No. 8. How does the grid of the lower tube in Fig. 21 receive its *signal voltage?*

No. 9. Why is it unnecessary to use a condenser for by-passing the bias resistor for self-biased push-pull class A tubes?

No. 10. What is the difference between an ordinary triode tube operated as a class B amplifier and one which is designed for this service exclusively?